D-TDR-64-674

(INTERIM REPORT)

SCIM -- A METER FOR MEASURING THE PERFORMANCE OF SPEECH COMMUNICATION SYSTEMS

TECHNICAL DOCUMENTARY REPORT NO. ESD-TDR-64-674

OCTOBER 1964

Karl D. Kryter
Jay H. Ball

DECISION SCIENCES LABORATORY
ELECTRONIC SYSTEMS DIVISION
AIR FORCE SYSTEMS COMMAND
UNITED STATES AIR FORCE
L. G. Hanscom Field, Bedford, Massachusetts

Project 2808, Task 280802

(Prepared under Contract No. AF 19 (628)-2463 by Bolt Beranek and Newman Inc., Cambridge, Massachusetts.)

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FOREWORD

This technical report presents the results of a study performed by Bolt, Beranek and Newman, Inc., Cambridge, Massachusetts, on Air Force Contract AF 19(628)-2463. This contract was initiated under Project 2808, "Psychoacoustic Standards in Voice Communication System Evaluation," Task 280802. This task was monitored under the direction of the Decision Sciences Laboratory, Mr. Stephen E. Stuntz, ESRHA.

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SCIM -- A METER FOR MEASURING THE PERFORMANCE OF SPEECH COMMUNICATION SYSTEMS

ABSTRACT

Two major theories have emerged that attempt to relate in a quantitative way the physical characteristics of a speech communication system and the intelligibility of speech as perceived by a crew of trained listeners.

The more widely used theory is the basis for the "Articulation Index" (AI) formulated in 1947 by N. R. French and J. C. Steinberg of the Bell Telephone Laboratories. This theory holds that the signal (speech)-to-noise ratio (in 20 narrow frequency bands) of a communication system will, when properly weighted and summed, provide an AI value that is directly related to the intelligibility of speech heard over that system.

A second theoretical approach to this problem is embodied in the Pattern Correspondence Index (PCI) machine proposed by J.C.R. Licklider in 1956. This instrument determines the correlation in the frequency and time domains between the speech input to a communication system and the speech output of that system.

The subject of the present paper is an electronic device called SCIM (Speech Communication Index Meter) that is designed primarily in terms of the Articulation Index theory; however, SCIM accommodates additional factors affecting speech that are not involved in the original procedures for calculating AI. The SCIM signal generator generates a 3-second signal burst, which is fed to the electrical input of the system under test; within 12 seconds the SCIM analyzer, located at the receiving station, calculates and displays an index, ranging from .00 to .99, which reflects the ability of that system to transmit intelligible speech.

Although devices based on similar principles have been previously developed, SCIM measures several parameters affecting the performance of a speech system differently than previous instruments. SCIM was designed to be relatively small, have small power requirements, be capable of remote digital readout, and contain other features to make it suitable for field, fixed station or laboratory use.

The results of SCIM measurements and actual speech intelligibility tests obtained from a variety of communication systems and operating conditions will be presented.

REVIEW AND APPROVAL

This technical report has been reviewed and is approved.

Joseph T. Begley
JOSEPH T. BEGLEY
Chief, Applications Division
Decision Sciences Laboratory

Roy Morgan
ROY MORGAN
Colonel, USAF
Director, Decision Sciences Laboratory

SECTION 1

INTRODUCTION

The valid assessment of the performance of speech communication systems has always been a difficult problem. The requirements for assessment have been, usually, two: (1) as aid and guidance in the design of electronic components (microphones, amplifiers, transmitters, receivers, earphones, etc.) and (2) the objective monitoring of the performance of communication systems during actual operations or under simulated operational conditions.

The task, of course, has been made difficult because, physically, the speech has such complex and dynamic characteristics, and secondly because the information content of the speech signal, as perceived by the human listener, is not always simply related to the physical nature of the speech signal, at least insofar as our present methods of physical analysis would indicate.

Speech intelligibility or speech articulation tests provide, when properly used, a reasonably accurate and valid procedure for measuring differences in the performance of communications equipment and systems. However, these psychological tests, which usually require laboratory conditions for their administration and specially trained crews of talkers and listeners, are costly and time-consuming, and, in most cases, cannot be readily applied to the measurement of speech systems under "real life" operational conditions.

Nevertheless, from 30 odd years of laboratory research with speech intelligibility and articulation tests, there has emerged a rather extensive set of facts about the relation between measurable physical characteristics of the received speech signal, including the noise, if any, present with the speech and the understandability of speech as perceived by listeners. Various investigators have attempted to use this knowledge to develop methods of system evaluation that avoid the speech testing process per se, but yet provide a measure that is interpretable in terms of speech understandability.

One outcome of this latter effort has been the development of methods for calculating from physical measurements of a communication system what the intelligibility of speech should be over that system; a secondary outcome has been the design of electronic test devices which, when applied to a speech communication system, automatically evaluate the performance of that speech system.

The purposes of the present paper are: (1) to outline briefly the theory and scientific basis for the design of one such test instrument (called the Speech Communication Index Meter, "SCIM") for measuring the performance of speech communication systems; (2) a description of the SCIM device; and (3) a comparison of the measurements achieved by the SCIM instrument for a variety of speech communication systems with the results of speech intelligibility tests obtained for the same systems.

SECTION 2

GENERAL RELATIONS BETWEEN INTELLIGIBILITY AND PHYSICAL CHARACTERISTICS OF SPEECH

Two major theories have emerged that attempt to relate in a quantitative way the physical characteristics of a speech communication system and the intelligibility of speech as perceived by a crew of trained listeners.

The more widely used theory is the basis for the "Articulation Index" (AI) formulated in 1947 by N. R. French and J. C. Steinberg[1] of the Bell Telephone Laboratories. This theory holds that the speech-to-noise ratio, in 20 narrow frequency bands, of a communication system will provide an AI value that is directly related to the intelligibility of speech heard over that system.

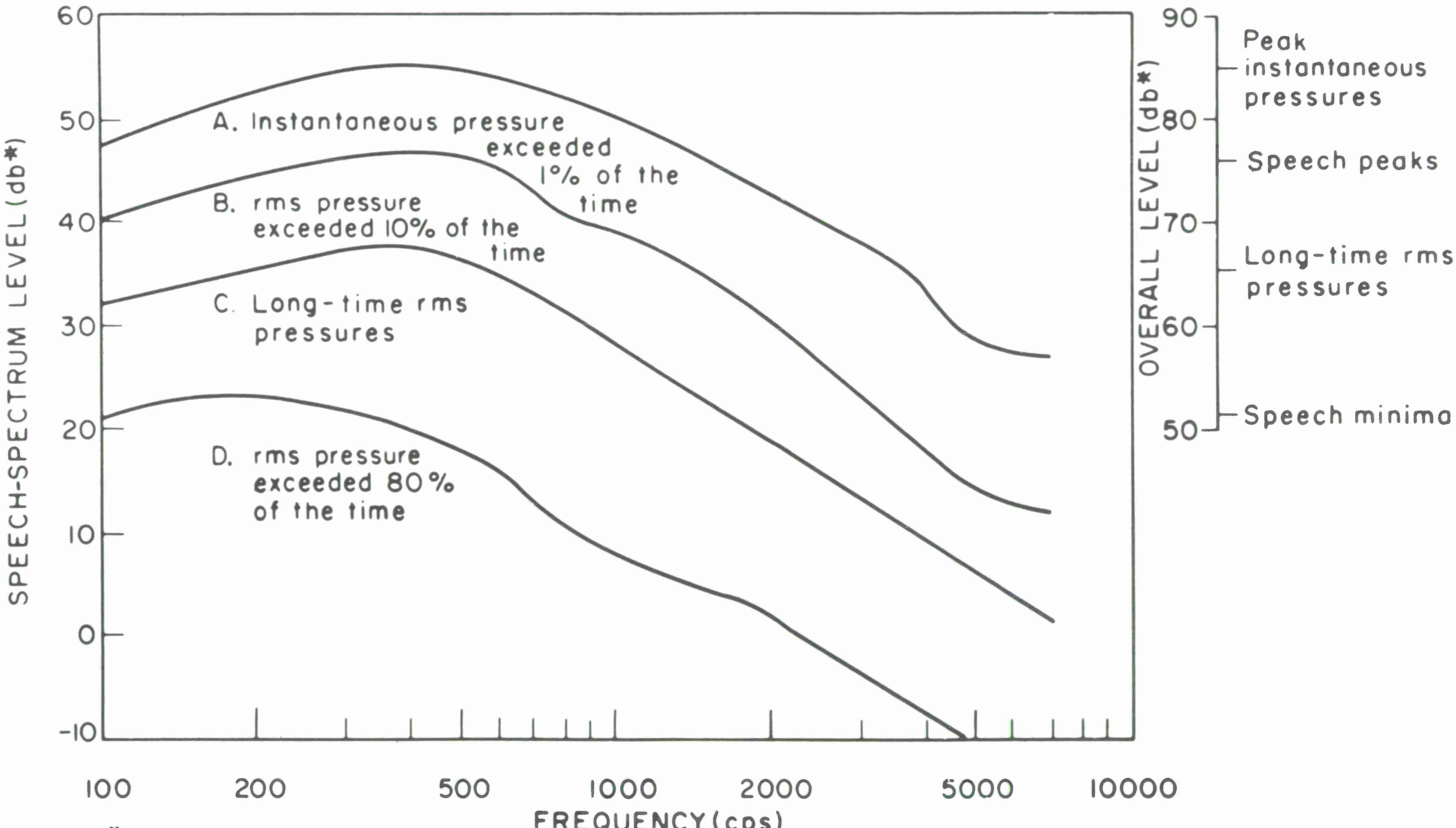
A second theoretical approach to this problem is embodied in the Pattern Correspondence Index (PCI) machine proposed by J. C. R. Licklider in 1956. This instrument determines the correlation in the frequency and time domains between the speech input to a communication system and the speech output of that system.

SCIM is designed primarily in terms of the Articulation Index theory; however, SCIM accommodates additional factors affecting speech that are not involved in the original procedures for calculating AI.

In order to explain the Articulation Index, it is necessary to describe some of the general characteristics of the speech signal and of masking by noise.

Figure 1 shows the variations in the pressure of conversational speech measured over short, 1/8-second intervals of time (1/8 second was chosen because it represents the average duration of individual speech sounds) as a function of frequency. The graph is plotted in terms of the pressure per cycle; we see that speech contains energy over the frequency range from at least 100 to 7000 cps and that the range of pressures is in excess of 30 dB over this entire frequency range. This range of pressures is the range found among individual speech sounds when the speech is uttered at a supposedly normal constant level of effort. It does not represent the range between weak talking or shouting, or amongst different talkers. This figure demonstrates the complex and dynamic nature of the speech signal.

<u>Masking</u>. A factor which enters into the calculation of AI is the masking effect of noise; noise containing sufficient energy, at a given point on the frequency scale, can mask or prevent the reception of speech components not only in the same frequency region, but also frequency components of speech that lie above and below the frequency components of the noise. Upward spread-of-masking, in which a low-frequency tone or noise masks a higher frequency portion of the speech spectrum, is far more severe than downward spread-of-masking, in which the masking tone is higher in frequency than the signal of interest. Figure 2 illustrates the effect of this "spread-of-masking" phenomenon upon the perception of pure tones.



*Re 0.0002 μbar

FIG. 1 SHOWING FOUR CURVES RELATING DIFFERENT MEASURES OF SPEECH LEVEL TO FREQUENCY (AFTER DUNN AND WHITE[5])

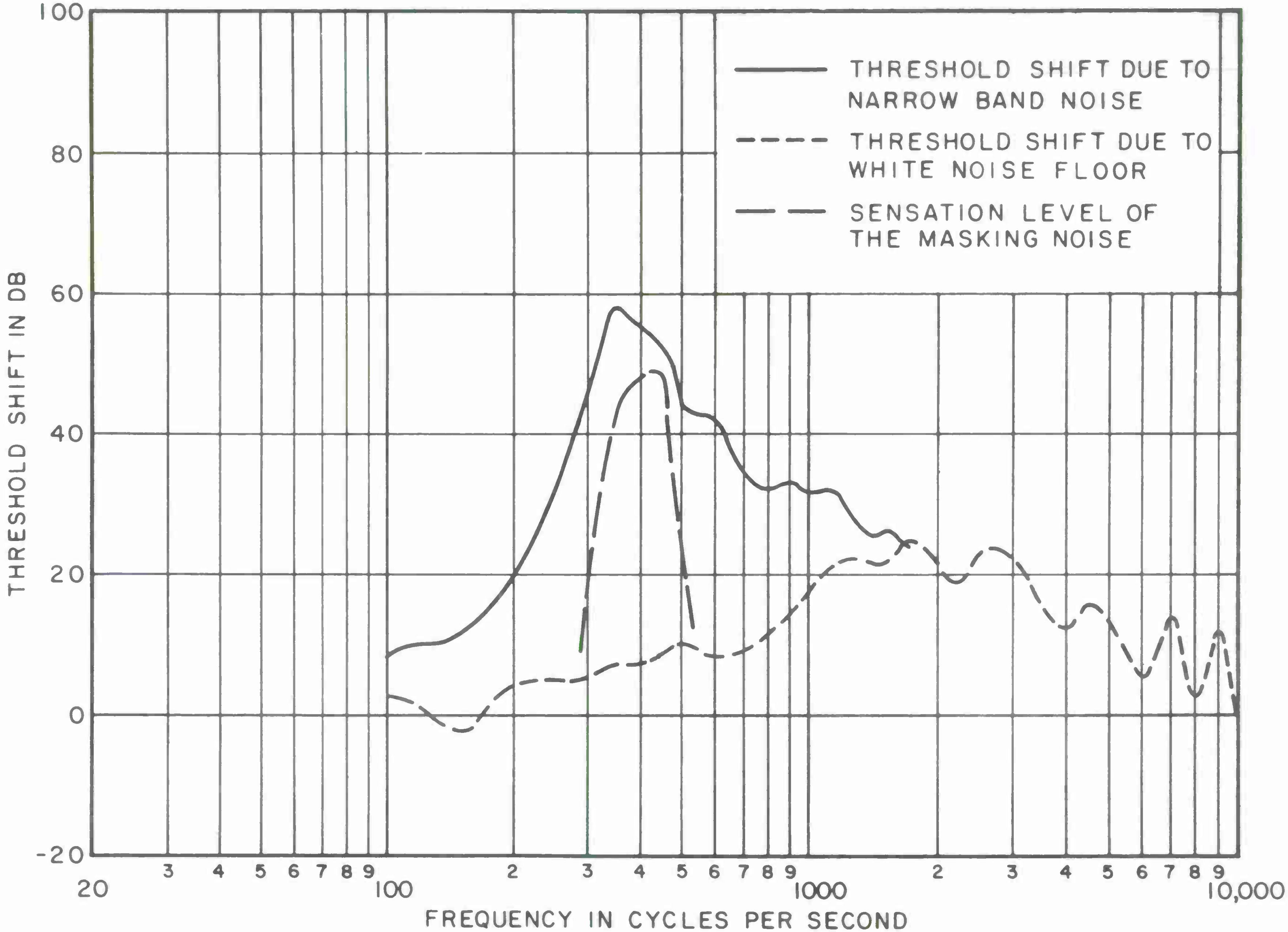


FIG. 2 THRESHOLD SHIFT DUE TO A 350-450 CPS MASKING NOISE AT 85 DB OVERALL SOUND PRESSURE LEVEL (AFTER CARTER AND KRYTER[6])

Figure 2 shows that a narrow band of noise centered at 400 cps would mask information around 800 cps as effectively as a band of noise centered at 800 cps but of 18 dB less amplitude than the band centered at 400 cps; it would also mask information at 1600 cps to much the same degree as would a -30 dB band of noise located at 1600 cps.

Calculation of the Articulation Index. As previously mentioned, the AI of a speech communications link is normally calculated by measuring the speech-to-noise (S/N) ratio in a number of frequency bands; the AI is defined as the weighted sum of these S/N ratios. There are two fundamental restrictions placed upon the calculated AI -- namely, that it can never be less than 0.00 and that it may never exceed 1.00. To implement the former restriction, we assume that all measured S/N ratios less than -12 dB are equal to -12 dB; and, since the band-weighting functions are derived so that the AI equals 1.00 when the S/N ratio in all bands is +18 dB, we assume that all S/N ratios in excess of +18 dB are equal to +18 dB. Figure 3 illustrates the general concept and work sheet utilized for the calculation of AI. In Fig. 3 the speech spectrum is divided into 20 narrow bands; these bands have been chosen so that each band contributes equally to speech intelligibility. Also, it is seen that speech at very weak and very intense levels does not contribute to intelligibility because it exceeds the limits of the normal ear.

On Fig. 3, we have sketched the speech spectrum that might be present at the listener's ear from a typical speech system along with an octave band of masking noise that is also presumed to be mixed with the speech at the listener's ear. Note that the masking spectrum, due to the aforementioned upward

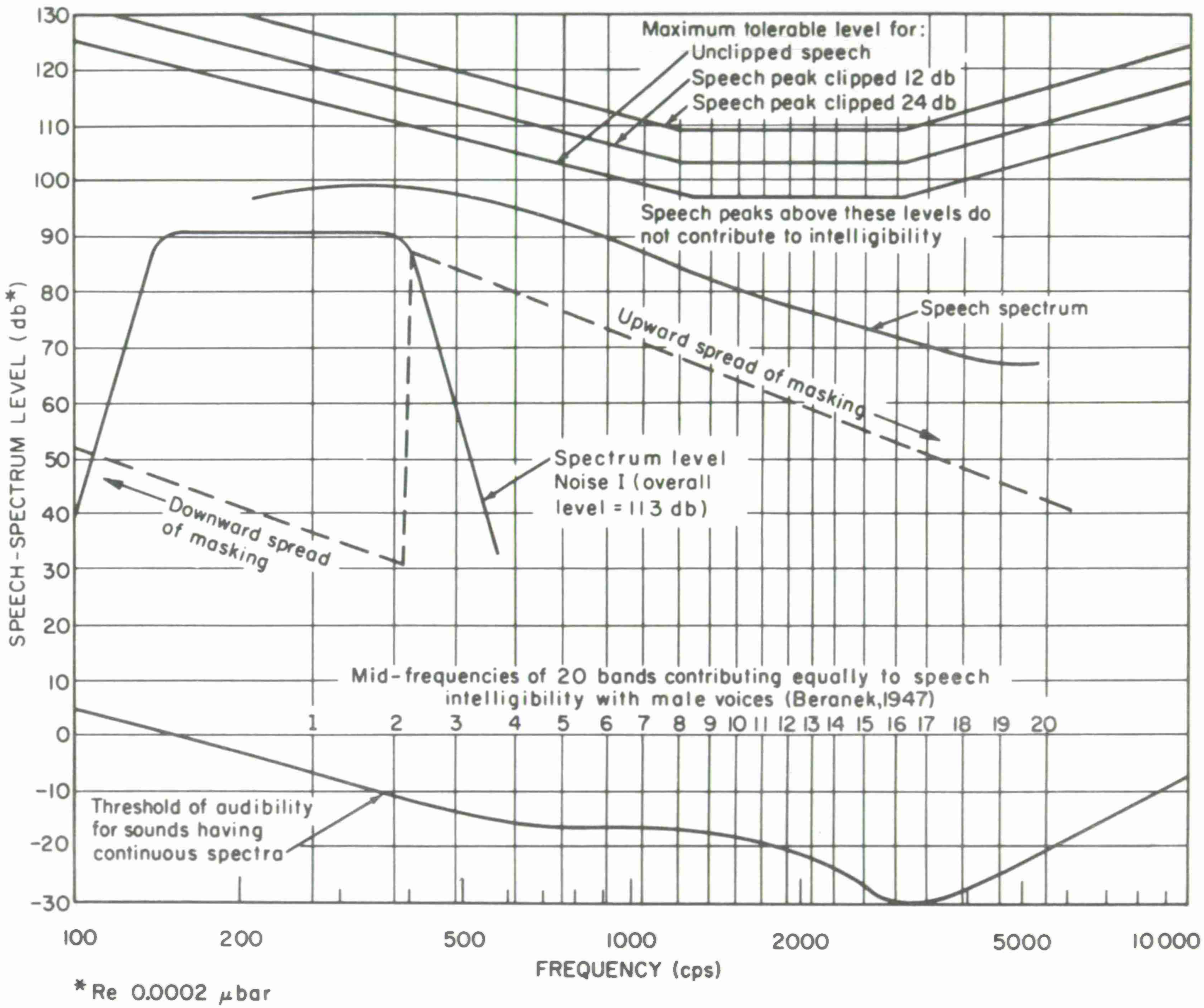


FIG. 3 WORK SHEET FOR THE CALCULATION OF THE ARTICULATION INDEX (AI). SPEECH SPECTRUM PLOTTED IS THE LONG-TERM RMS OF SPEECH PLUS 12DB, THE DASHED CURVE IS THE SPREAD-OF-MASKING TO BE EXPECTED FROM THE OCTAVE BAND OF NOISE THAT IS PLOTTED ON THE CURVE (SEE REF.2).

spread-of-masking of the noise exceeds, at the higher frequencies the actual noise spectrum. The AI is calculated[2] by taking the S/N ratios between the plotted speech spectrum (long-term rms plus 12 dB) and the noise or masking spectrum, whichever is the greater for each of the 20 bands, weighting proportionately each S/N ratio so that +30 dB equals 1.00 and 0 dB equals 0, summing the result and dividing by 20.

Frequency shift. The AI must be corrected for factors other than noise that affect intelligibility. One such factor is frequency shift. If a single sideband ratio receiver is not tuned precisely to the carrier frequency of the transmitter, the detected speech signal will appear shifted upwards or downwards by an amount equal to the mismatch between the transmitted carrier frequency and the receiver local oscillator frequency. While upward shifts in the speech spectrum of 100 cps or more can be tolerated, a downward shift of 50 cps can have a devastating effect upon the intelligibility of the speech.

The correction factor to be applied because of the presence of frequency shift will be shown in a later figure.

Peak clipping. Peak clipping is still another factor which enters into the calculation of AI. Figure 4 shows the increase (over unclipped speech of the same peak amplitude) in the long-term rms pressure of peak-clipped speech as a function of the amount of clipping and subsequent reamplification

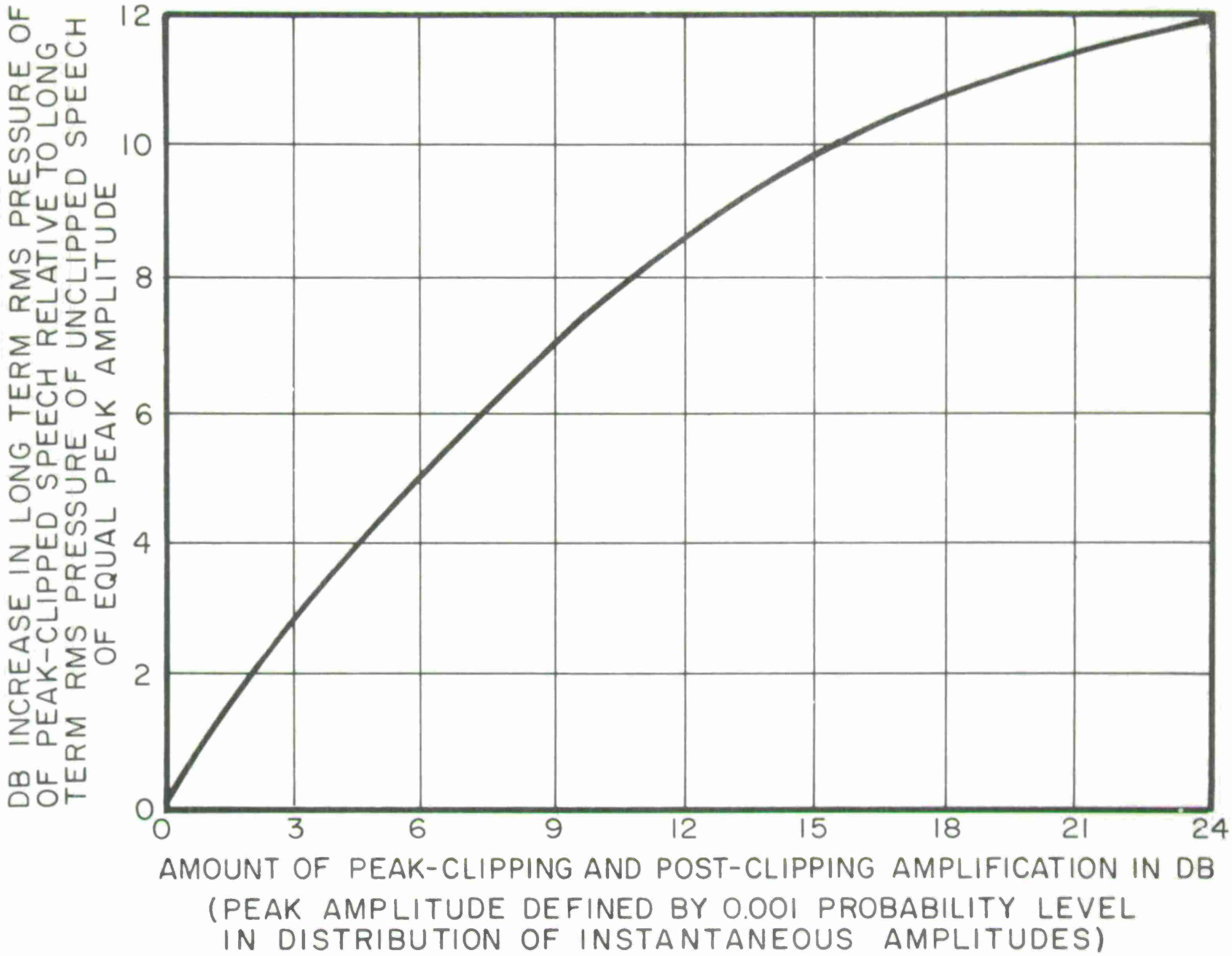


FIG. 4 SHOWING THE INCREASE IN RMS SPEECH POWER AS A FUNCTION OF CLIPPING WHEN CLIPPED LEVEL IS RAISED TO CLIPPING REFERENCE LEVEL (AFTER WATHEN-DUNN AND LIPKE[7])

required to restore the unclipped peak levels. Peak clipping must be taken into account when calculating AI because the intelligibility of speech is, to a first approximation, proportional to the long-term rms signal-to-noise ratio and not to the peak S/N ratio.

Validity of AI. The validity of the AI concept has been well established for a wide variety of speech communication systems. Some of the results of the application of the AI calculation procedure to speech systems for which speech intelligibility test scores were available are presented in Figs. 5 and 6. Figure 7 shows the relation between AI and different types of speech test materials. It is to be noted that the percentage of test items correctly perceived is dependent not only on the speech material -- nonsense syllables, words, or sentences -- but also the size of the message set, whether, for example, the test vocabulary consisted of a 1000, 256, or 32 "PB" (so-called phonetically balanced) words.[3]

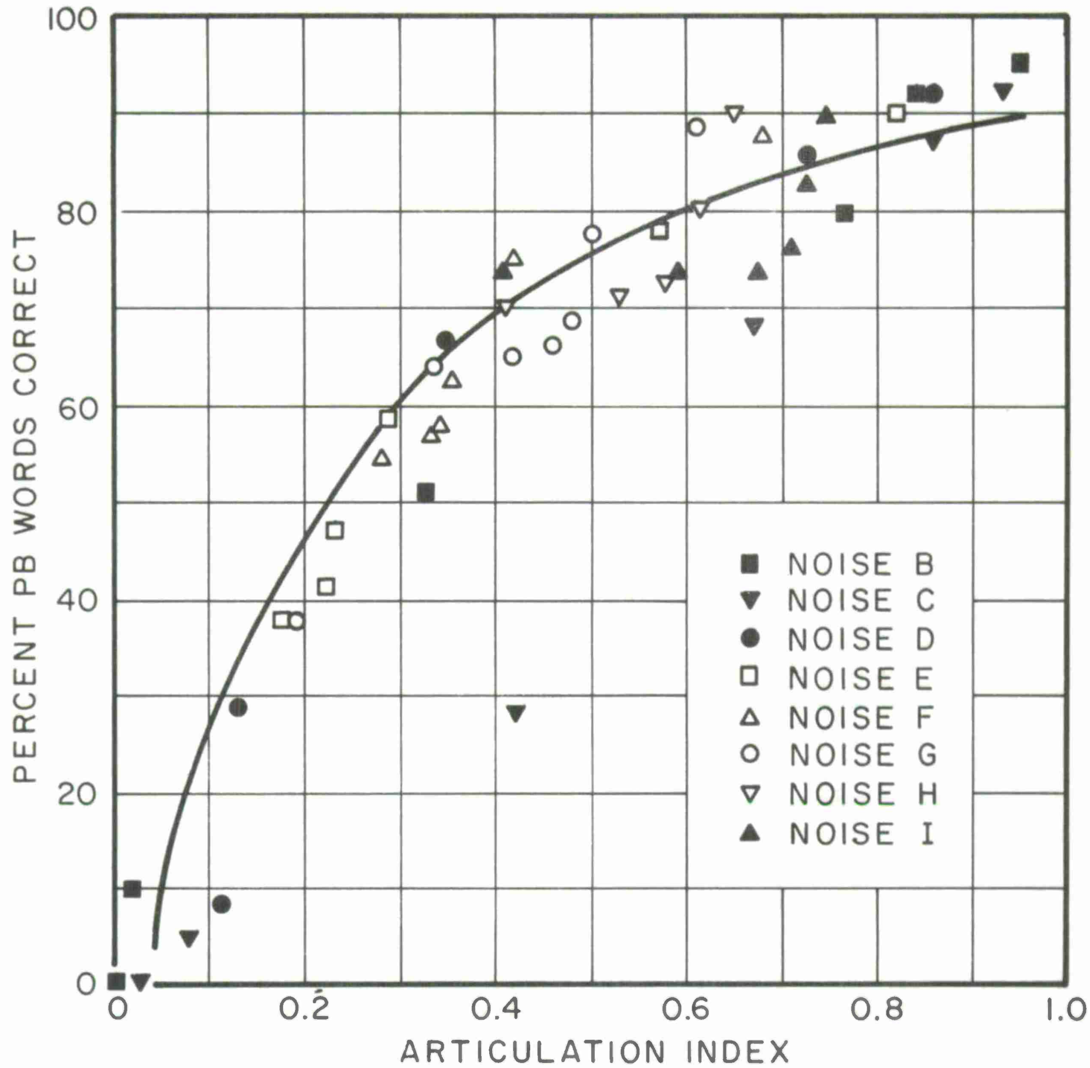


FIG. 5 COMPARISON OF OBTAINED AND PREDICTED TEST SCORES FOR BROAD-BAND SPEECH IN THE PRESENCE OF NARROW BANDS OF NOISE SET AT VARIOUS INTENSITY LEVELS (AFTER MILLER[8])

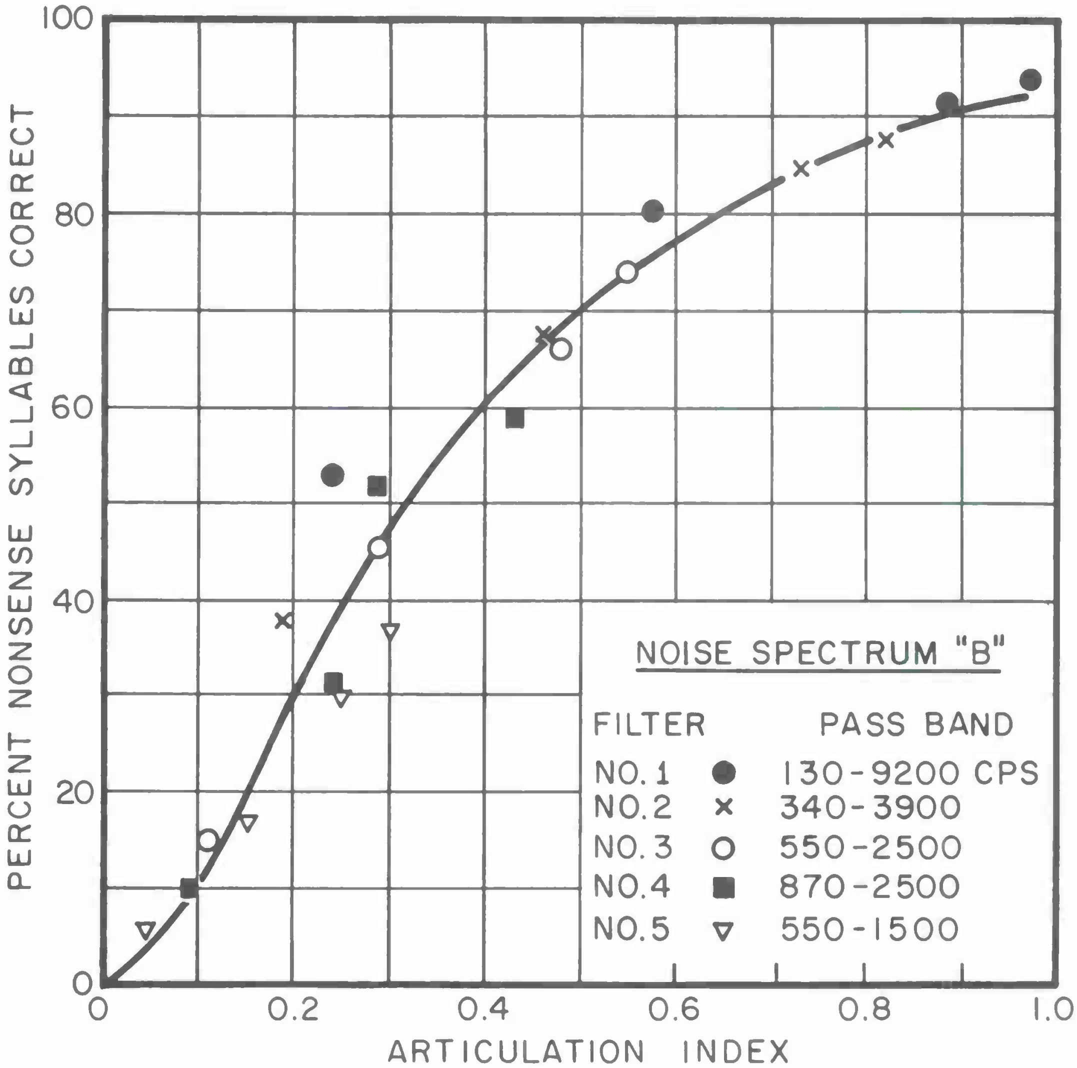


FIG. 6 COMPARISON OF OBTAINED AND PREDICTED TEST SCORES FOR SPEECH PASSED THROUGH A BANDPASS FILTER AND HEARD IN THE PRESENCE OF A BROAD-BAND NEGATIVELY SLOPED SPECTRUM NOISE SET AT VARIOUS INTENSITY LEVELS (AFTER EGAN AND WIENER[9])

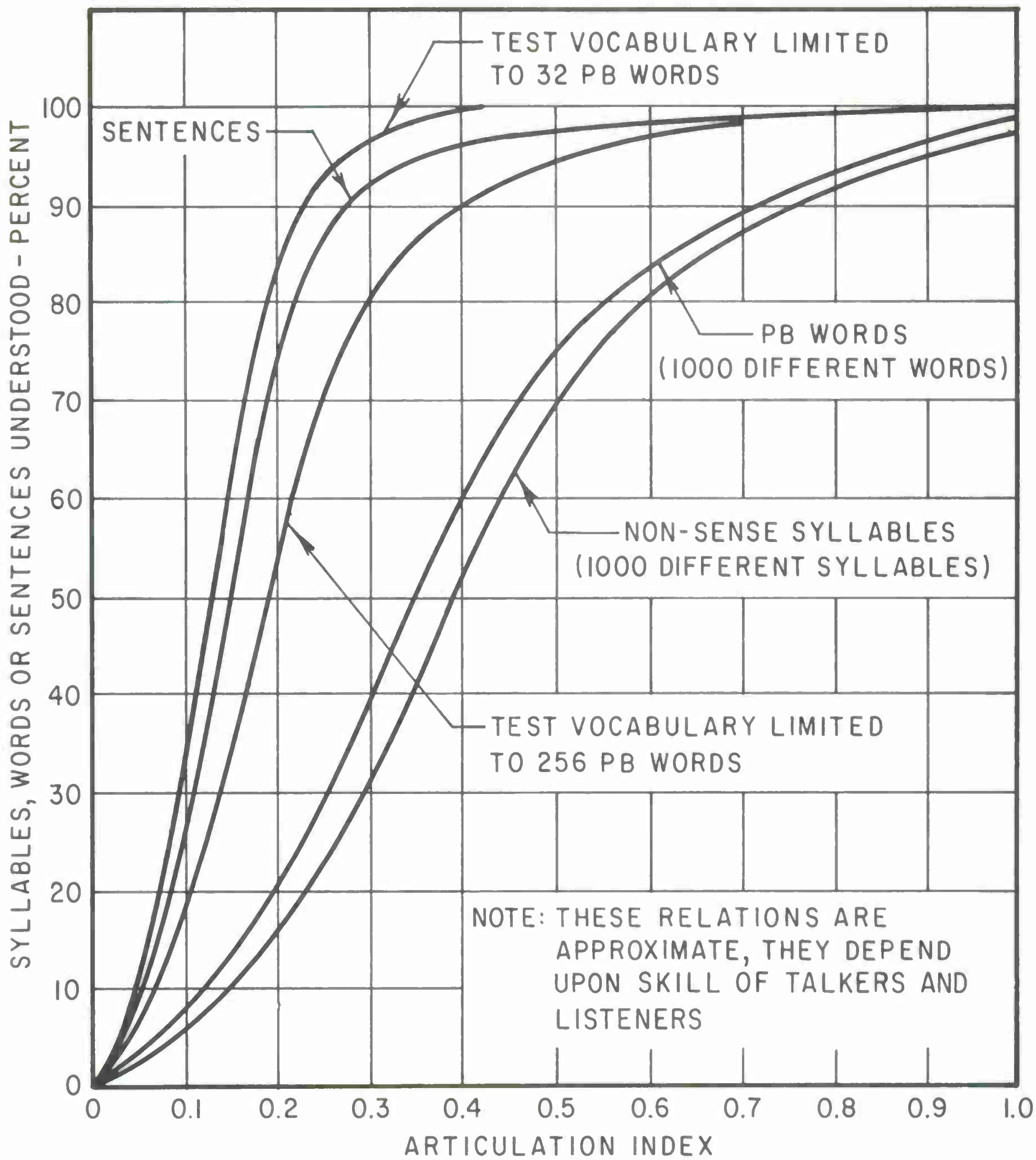


FIG. 7 RELATION BETWEEN AI AND VARIOUS MEASURES OF SPEECH INTELLIGIBILITY (AFTER FRENCH AND STEINBERG[1] AND MILLER[8])

SECTION 3

SCIM

Although the AI procedure is objective, it requires tedious calculations from physical data about speech systems that are not always easily obtained.

The Speech Communication Index Meter is an electronic device which automatically calculates, albeit in a somewhat simplified fashion, the AI of any given speech system. There has been at least one previous instrument built on similar principles. General Electronics Laboratories Inc. of Cambridge, Massachusetts, designed and built for the U. S. Army an instrument called the Voice Interference Analysis Set (VIAS) which also estimates the AI of speech systems. VIAS differs, however, in a number of respects from SCIM, both in the type of signals utilized for system testing and in the processing and analysis of these signals.

Signal generator. The SCIM signal generator, shown in Figs. 8 and 9, located at the transmitting terminal of the link under test normally presents for transmission a continuous 1 kc calibration tone. When the START button is depressed, the signal generator synchronizes itself, resulting in a maximum delay of one second before the beginning of the test sequence shown in Fig. 10.

The sequence begins with the replacement of the 1000 cps calibration tone by a 30-millisecond sync burst. The burst consists of the sum of a 600 cps sine wave and a 2000 cps sine wave; the function of the burst is, of course, to signal the analyzer that the test sequence has begun.

Following the sync burst there appears a "speech" signal, produced by square-wave modulating a shaped-spectrum random noise

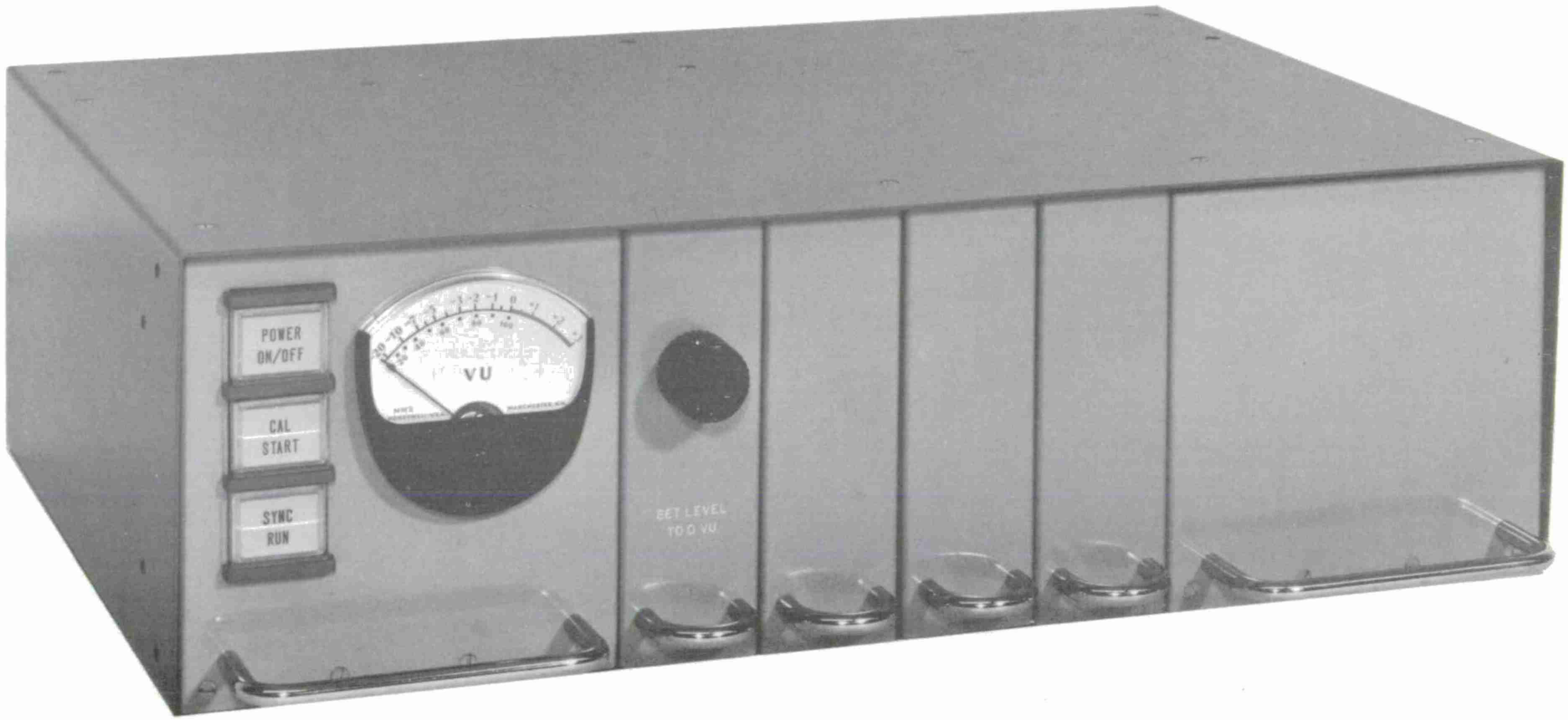


FIG. 8 PHOTO OF SIGNAL GENERATOR

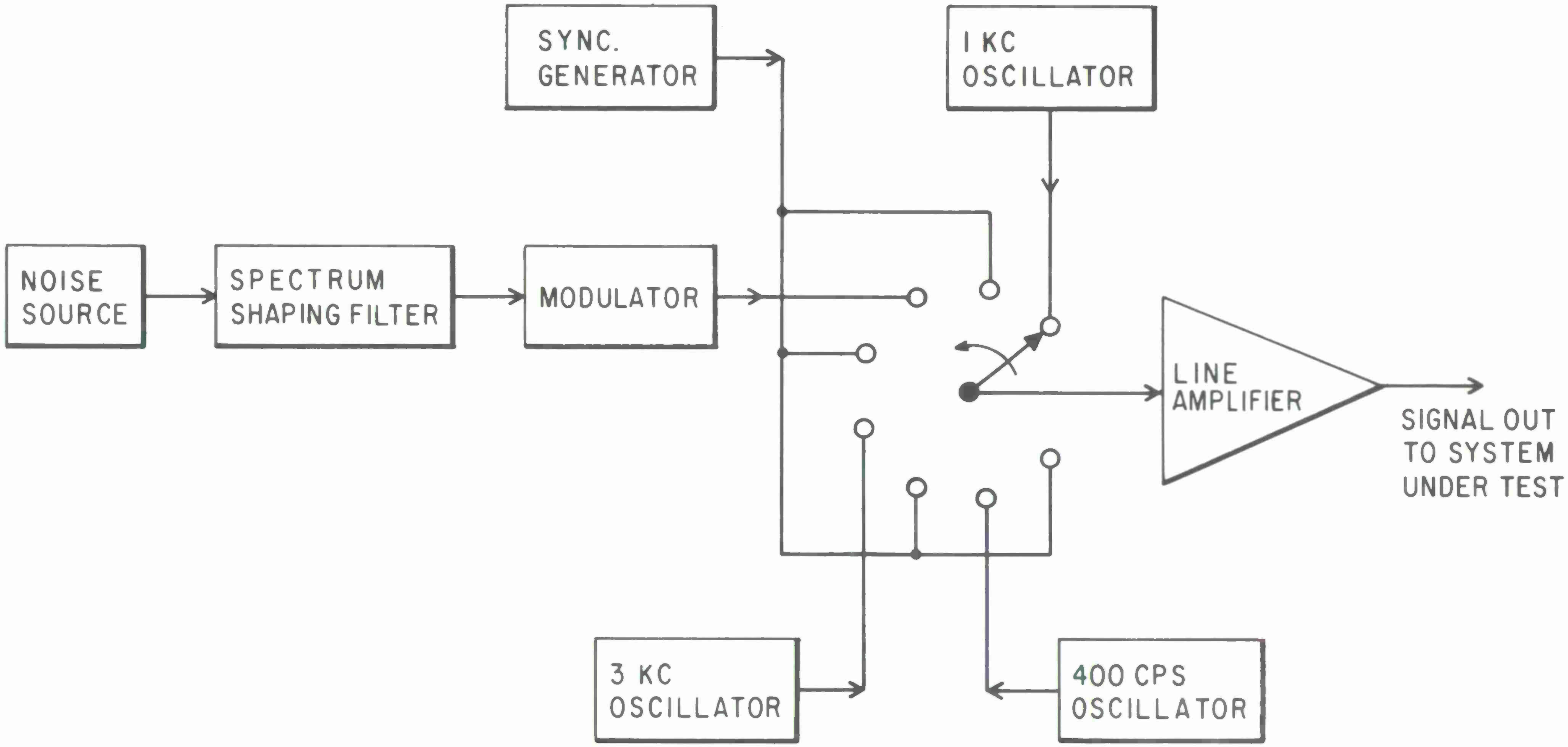


FIG. 9 SIMPLIFIED BLOCK DIAGRAM OF SIGNAL GENERATOR

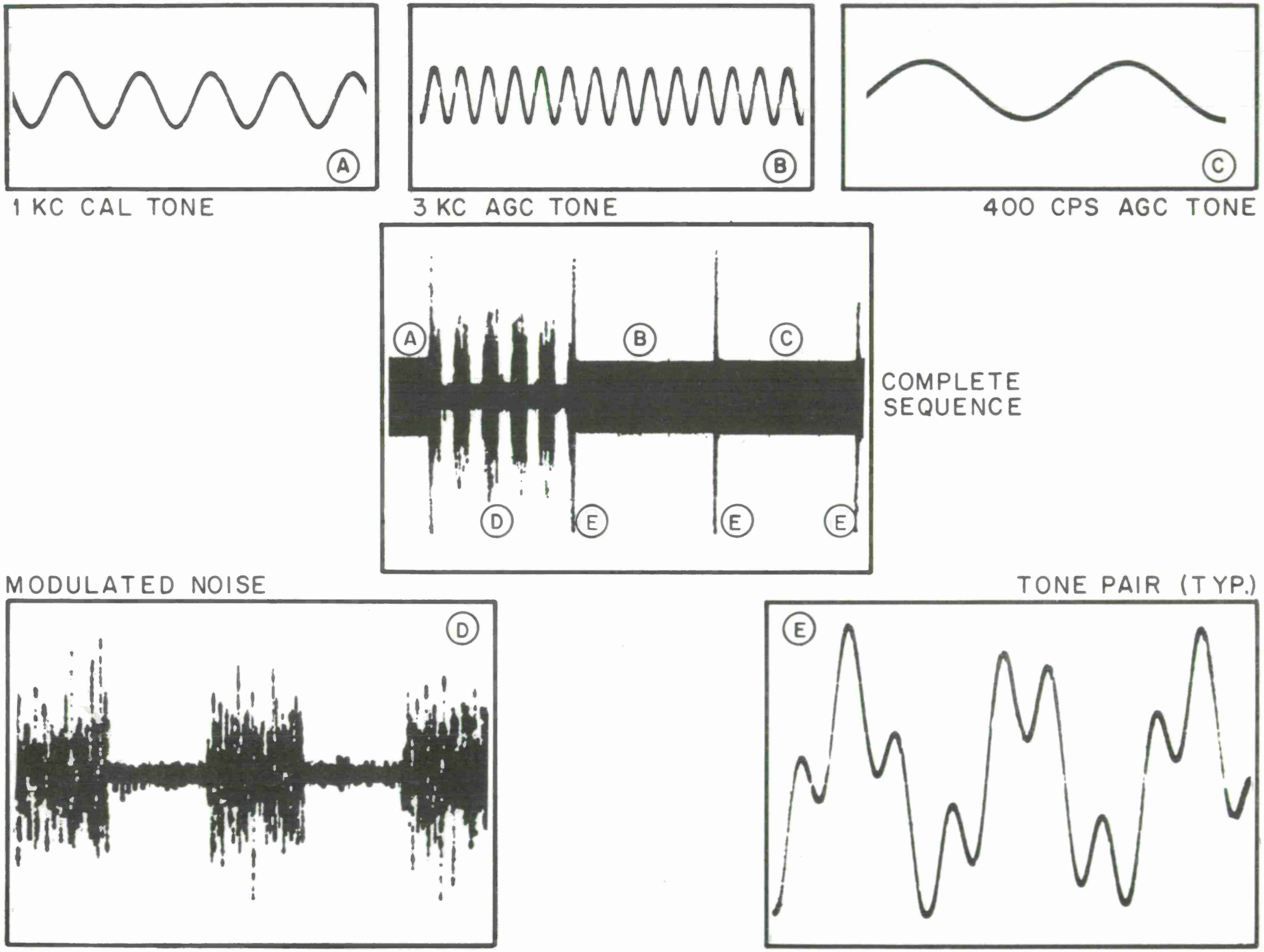


FIG. 10 TEST SIGNAL SEQUENCE

source. Prior to modulation the noise is tailored, by means of a spectrum-shaping filter, to have an amplitude-versus-frequency characteristic which closely approximates the long-term root-mean-square spectrum of speech. Modulation is performed at a 5 cps rate by means of a time-varying attenuator which provides an on-to-off attenuation ratio of 17 dB. Since the modulation duty cycle is 50%, the high-level and low-level portions of the "speech" signal each last for 1/10 second.

At the end of one second, the "speech" signal transmission is terminated and a 3000 cps tone appears instead. A 30-millisecond sync burst is used to mark the temporal boundary between the "speech" and the tone. While the 3000 cps tone is on, measurements are made by the analyzer of the background noise, in the frequency range 250-1650 cps, of the communication system under test; the tone serves to keep constant the gain of any AGC equipment which may be part of the system being tested. After one second has elapsed, the 3000 cps tone is replaced by a 400 cps tone, which serves the same purpose but allows system noise measurements to be made in the 1650-4200 cps portion of the spectrum.

As before, a 30-millisecond sync burst delimits the end and beginning, respectively, of the 3000 and 400 cps AGC tones. At the end of this third one-second interval a sync burst is presented and the signal generator output reverts to the 1000 cps calibration tone, which will continue until the next time that the START button is depressed.

Analyzer. Located at the receiving terminal, the SCIM analyzer (shown in Figs. 11 and 12) filters the received "speech" or noise spectrum into nine bands, which together cover the frequency range from 250 cps to 4200 cps. The lower-skirt slope of all filters is +18 dB/octave, the upper-skirt -60 or more

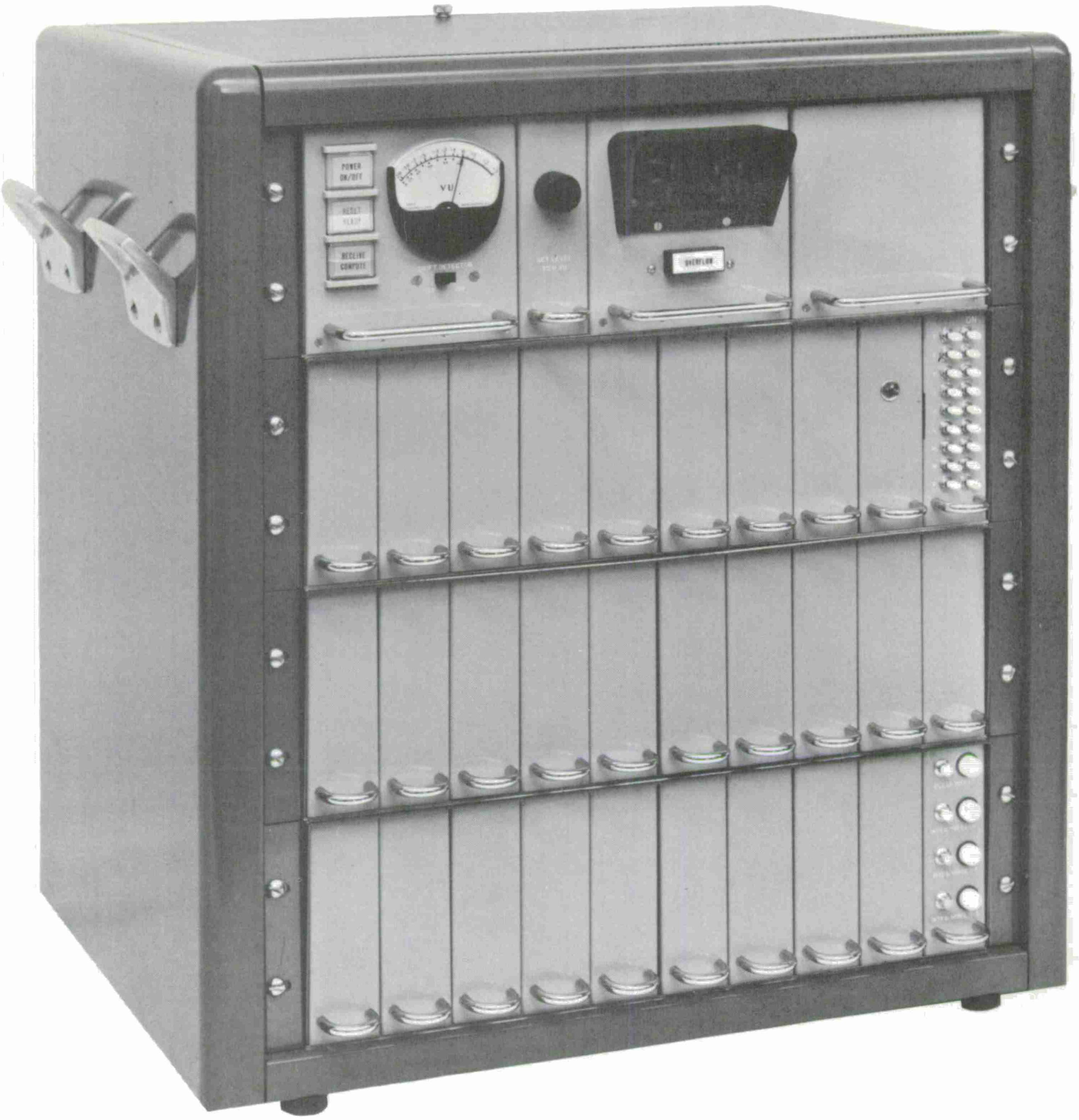


FIG. 11   PHOTO OF ANALYZER

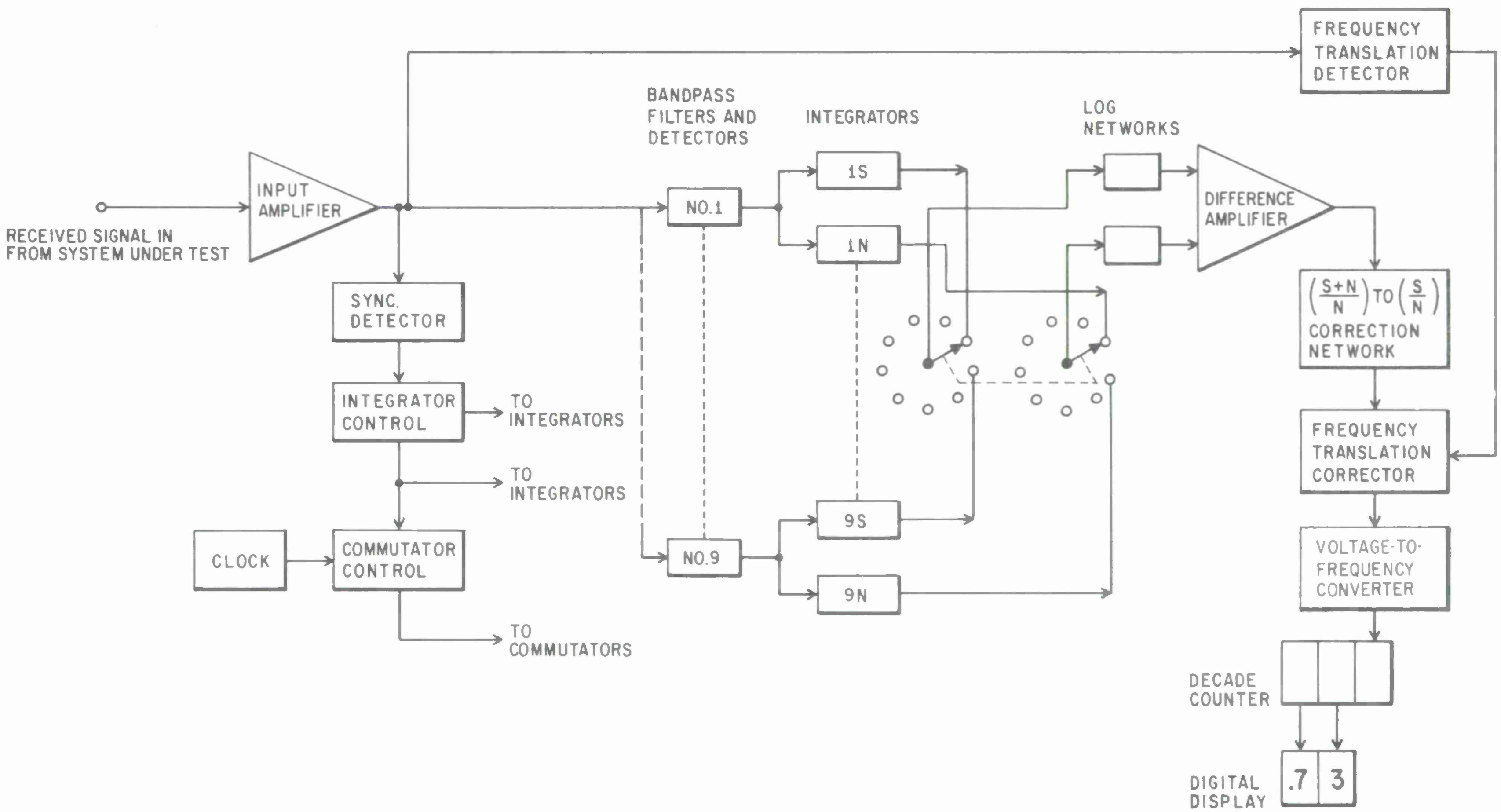


FIG. 12 SIMPLIFIED BLOCK DIAGRAM OF ANALYZER

dB per octave; these slopes were chosen specifically to approximate the inverse of the idealized spread-of-masking curve.*

The output of each filter is detected and fed to a "speech" and a noise integrator. The integrators are of the operational-amplifier/feedback-capacitor type, and provide true time integration of the unipolar detector output signals. The "integrate/hold" and "clear" functions are controlled by reed relays mounted directly on each integrator printed circuit board.

The "speech" integrators are enabled during the one-second interval in which the "speech" signal is produced by the signal generator. At the end of this one-second period, integration ceases and the final value of the integral is held. The value stored in the "speech" integrators, it should be noted, is really the integral of the detected "speech" plus system noise.

During the following one-second interval, the noise integrators for the first five bands are enabled. Since no "speech" is transmitted at this time, the final voltage stored in these integrators is proportional to the noise alone. The frequency of the AGC tone during this interval, the reader will recall, is 3000 cps; this frequency is greatly attenuated by all of the filters in bands 1 through 5, so that the presence of the tone does not affect the measurements made in these bands.

At the end of this second one-second interval, the band 1-5 noise integrators are switched to the "hold" mode, and in the

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* The spread-of-masking function is dependent upon both the frequency and level of the masking noise; for the purposes of SCIM we have chosen an approximation which in our opinion, is suitable for use with typical communications systems. This idealized spread-of-masking curve falls off above the frequency of the masking noise at a rate of 18 dB per octave; the masking effect at frequencies below that of the masking noise is considered to be negligible.

third one-second interval the noise integrators associated with bands 6 through 9 are enabled.

After the entire 3-second sequence has been completed, the analyzer enters its "calculating" mode. At this time, the outputs of the band 1 (250-500 cps) "speech" and noise integrators are connected, by means of a commutating switch, to two logarithmic converters. Each log converter is a temperature-stabilized diode network which delivers, at its output, a d.c. voltage proportional to the logarithm of the d.c. voltage of its input.

The outputs of the two log networks are compared by a very stable d.c. differential amplifier. The output of this amplifier is proportional to (log["speech" + noise]) - (log[noise]), which may be rewritten as

$$\log \left[\frac{\text{"speech"} + \text{noise}}{\text{noise}} = \log \frac{S + N}{N} \right] .$$

A diode correction network, having the transfer characteristic

$$e_{out} = K \log 4 \left(10^{e_{in}} - 1\right)$$

effects the transformation of log (S+N/N) into log (S/N) + 12 dB, as required by the method for AI calculation.

The d.c. output voltage of the diode network is fed via a unity-gain buffer amplifier to a voltage-to-frequency converter. This device accepts a d.c. voltage at its input and outputs a pulse train, the PRF (pulse repetition frequency) of which is directly proportional to the applied voltage. Thus, <u>in a fixed time</u>, the converter generates a <u>number</u> of pulses directly proportional to the S/N ratio. These pulses are counted by a three-place decade counter whose displays (of the two higher-order digits only) constitute the readout display of the analyzer.

After dwelling for one second on the outputs of the band 1 "speech" and noise integrators, the commutator is advanced one step, thereby connecting the outputs of the corresponding band 2 integrators to the two log converters. After another one-second dwell interval has elapsed (during which time the decade counter adds to its previous total a number proportional to the S/N ratio in band 2) the commutator examines the band 3 integrators, etc. When the commutator comes to rest in position 10, the S/N ratios of all nine analyzer bands have been totalized; the number displayed is the Speech Communication Index (SCI), and is defined as

$$SCI = \sum_{i=1}^{9} [20 \log (S/N)_i + 12 \text{ dB}] \quad .$$

The theory behind the Articulation Index requires that, in the calculation of AI, all S/N ratios in excess of +18 dB be considered equal to +18 dB. The reason for this boundary condition is simply that the contribution to intelligibility of any given band improves as the S/N ratio improves <u>and reaches its maximum value at an S/N ratio of +18 dB</u>. Larger S/N ratios, therefore, cannot further improve the intelligibility, and should not contribute to the total AI in an amount greater than that corresponding to a +18 dB S/N ratio.

The AI theory also requires that S/N ratios less than -12 dB (corresponding to zero contribution to intelligibility) be considered equal to -12 dB. The reasoning is, of course, just the converse of the argument given above for the limiting of S/N ratios in excess of +18 dB.

The boundary conditions described above are imposed in the analyzer by the d.c. differential amplifier. Its circuitry is such that its output cannot rise above +18 volts. Since this output level has a scale factor of 1 dB/volt, an 18 volt output corresponds to an 18 dB S+N/N [or (S/N)] ratio; after the "addition" of 12 dB by the diode correction network, this corresponds to the upper limit of the required 30 dB range.

The amplifier also serves to impose the lower boundary limit; since its output must always be $\geq$ 0 volts (i.e., $\geq$ 0 dB).

A complete measurement run, including three seconds of "speech" and AGC tone transmissions and nine seconds of "calculation" time may take as little as 11 seconds, since the first second of calculation time can be coincident with the third second of transmission from the signal generator.

Analyzer sync detector. The analyzer sync detection system consists of two filters, centered at 600 cps and 2 kc, respectively, whose outputs are detected and ANDed together. Whenever both sync tones are found to be present, a relay is actuated which advances a cam-operated rotary stepping switch. This switch, in turn, provides sequentially the commands for the analyzer to clear all integrators, integrate "speech," integrate noise, begin calculation, etc.

The analyzer control system also contains a timing mechanism for the commutator, which sequentially connects the logarithmic attenuator inputs to the outputs of successive pairs of integrators.

Prior to an actual measurement run, the frequency of the 1 kc calibration tone is measured by means of a special discriminator filter. The filter output is detected and compared with the detected input to the discriminator. The difference between these two d.c. voltages is a measure of the mismatch between the transmitter carrier frequency and the receiver local oscillator frequency.

A d.c. servo loop utilizes this "tuning error" voltage and sets a motor-driven servo-potentiometer according to the degree of tuning error (frequency shift) present. The servo-potentiometer attenuates the d.c. voltage fed to the voltage-to-frequency converter, and thereby directly multiplies the AI by some number $\leq$ 1.00. The shape of the discriminator filter is adjusted so that the resulting decrement in AI is in accordance with the data in Fig. 13.

Operator requirements. Ease of operation of the SCIM, in the field and by untrained personnel, has been given major consideration; the only front-panel controls on the signal generator are a power on/off switch, a metered level control, and a press-to-start switch. The generator is internally calibrated so that, once the level of the 1000 cps calibration tone is established and the START button is pressed, the various signal sources within the generator will be presented to the input of the communication link under test in the proper sequence, for the proper durations and at the required levels with no further attention from the operator.

The controls of the analyzer are equally simple - a power on/off switch, a metered level control and a manual reset switch. The operator need only set the level of the incoming 1000 cps calibration tone to 0 VU on the meter. The manual reset switch need be used only after power is first applied to the analyzer or when a severe signal dropout results in loss of synchronization during a measurement run. Under normal circumstances the analyzer will be ready for a second measurement run immediately upon completion of the first. It will continue to display its last calculated AI indefinitely, unless power is removed or another run is begun. The operator may, therefore, read the calculated AI from the digital display at his leisure.

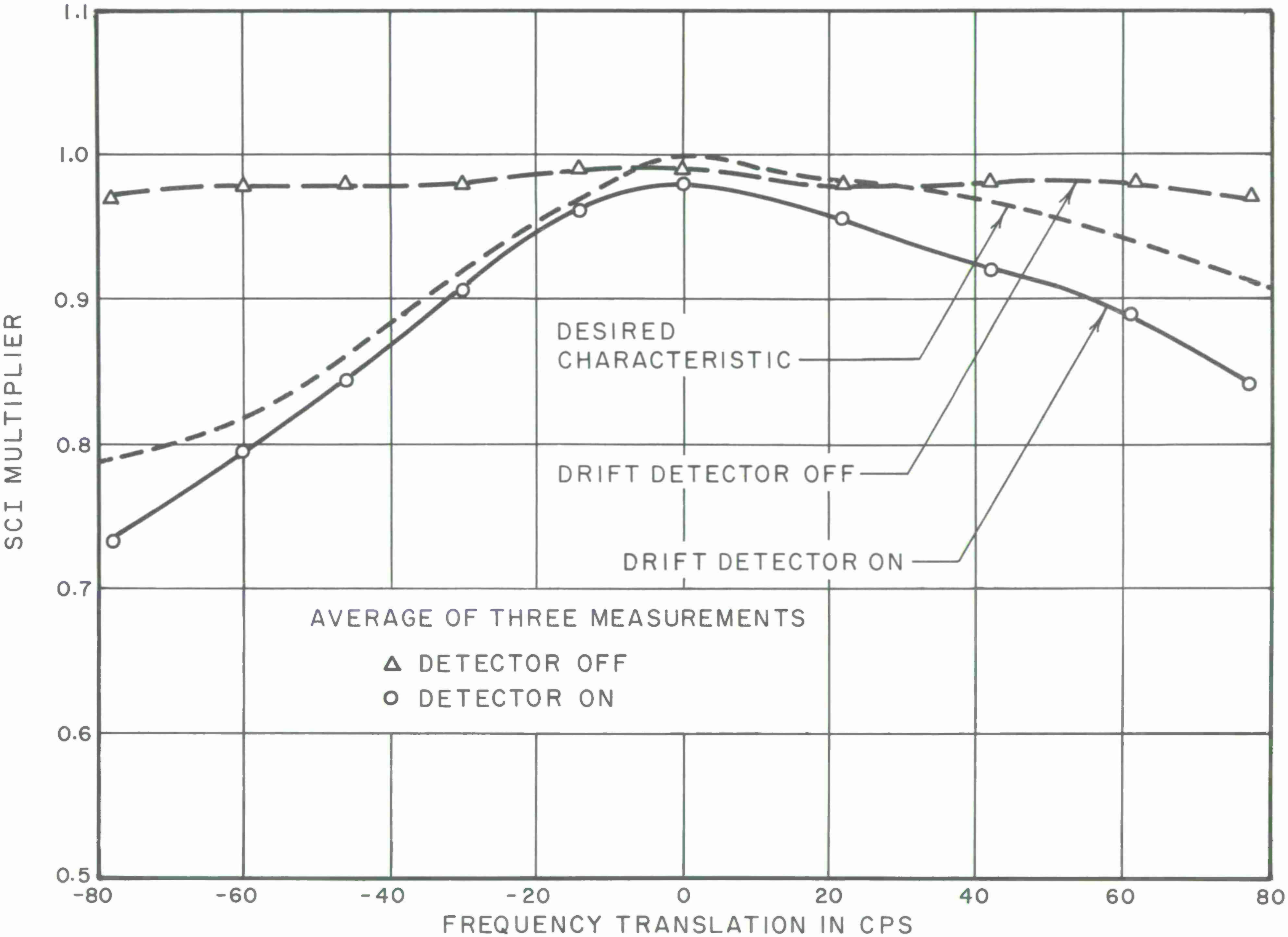

FIG. 13 SCI MULTIPLIER vs. FREQUENCY SHIFT

SECTION 4

TEST RESULTS

There are two criteria that can be used for evaluating the accuracy and reliability of SCIM -- (1) how well "SCI's" for a variety of speech systems agree with AI's calculated in accordance with the 20-band method for calculated AI; (2) how well the SCI's, for the systems being evaluated, predict the intelligibility of speech as measured with typical psychoacoustic test methods.

SCI vs AI. Figure 14 illustrates the agreement between calculated AI's and SCI's as measured by SCIM when the S/N ratio present in an otherwise "high-fidelity" speech system was systematically varied. It is seen in this figure that except for values below about 0.4, SCIM estimates the calculated AI within $\pm$ .02 of its true value; below 0.4 the disagreement increases to approximately $\pm$ .04. Thus, for at least the system and noise conditions represented in Fig. 14, an average of one, or perhaps two, readings of SCIM would provide an SCI that has nearly exactly the same AI as would be found by the standard calculation procedures.

SCI vs speech test scores. A series of psychoacoustic tests were conducted on a wide variety of speech systems in order to determine the relation between SCI and measured speech intelligibility. For these tests a listening crew of eight high school and college students were used. The speech tests, "Modified Rhyme" (MR)[4] were recorded at the output of the various systems and noise conditions to be evaluated; these recordings were then administered to the listening crew in

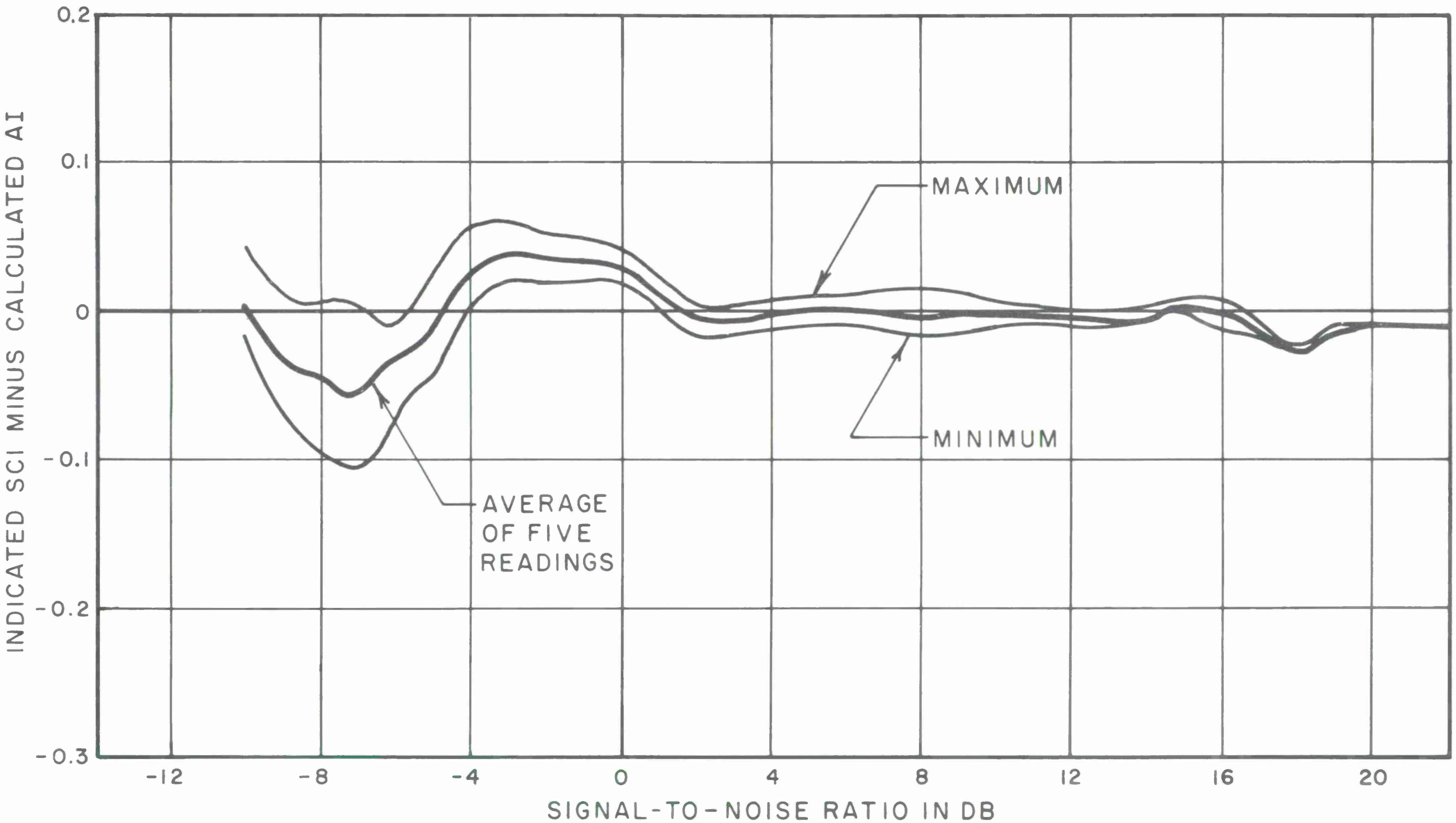

FIG. 14 SHOWING THE MAXIMUM, MINIMUM AND AVERAGE VALUES OF THE DEVIATION OF FIVE MEASURED SCI'S FROM THE CALCULATED AI'S AT S/N RATIOS FROM -10 DB TO +22 DB

accordance with an experimental design that more or less randomized the order in which the different systems and conditions were tested. Two 50-word MR for each of two male talkers were given at each test condition. The listeners wore earphones and were seated in a soundproofed room. All noise conditions were achieved by electrically mixing, at the output of the speech transmission system, the speech with a white noise that had been passed through a 4400 cps low-pass filter. The signal-to-noise ratios (S/N) were measured on a VU meter. The peak VU meter readings, in the absence of the masking noise, were averaged for a sample of the test words. The speech signal was then turned off and the level of the noise adjusted until the VU meter had the same value as the average found for the peaks of speech in the absence of the noise; the S/N was then said to be 0 dB, if the noise exceeded the average speech peaks by 5 dB, the S/N was called -5 dB, and so forth.

Five SCI's were measured with SCIM for each of the test conditions. The results of the speech tests and the SCI measurements are presented in Table 1, and the averages, for most of these results, in Fig. 15.

Figure 15 shows that (except for certain of the 75 cps frequency shift with peak clipping conditions) the measured SCI values predict the test scores within ± about 10 percentage points. This amount of deviation (when leaving out some of the 75 cps frequency shift conditions) compares reasonably well with the range of differences between calculated AI and measured intelligibility, as shown in previous studies given in Figs. 5 and 6.

TABLE 1

Condition Number	S/N, db Note 1	Frequency Shift, cps	Peak Clipping, dB	Low-Pass Filter Cutoff Frequency Kc - Note 2	Average of 2 MR Tests Talker CW	Average of 2 MR Tests Talker AH	Average of 4 MR Tests Two Talkers	Average of 5 SCI Readings	Maximum Deviation of any SCI Reading from Average
1	+15	0	0	5	94.5	79.4	87.0	.71	±.01
2	+ 3	0	0	5	74.0	66.0	70.0	.35	+.01
3	- 3	0	0	5	50.6	53.1	51.8	.27	+.01
4	0	0	0	5	96.7	88.3	92.5	.84	+.08
5	+15	0	0	3	86.0	75.0	80.5	.79	+.01
6	+15	0	0	1.5	73.2	70.3	71.8	.57	±.01
7	+15	+75	0	5	93.7	78.7	86.2	.64	+.01
8	+15	-75	0	5	91.0	77.3	84.1	.44	-.03
9	+ 3	0	0	3	65.7	63.7	64.7	.37	+.02
10	+ 3	0	0	1.5	69.0	55.8	62.4	.39	+.02
11	+ 3	+75	0	3	63.7	64.6	64.2	.46	±.01
12	+ 3	+75	0	1.5	69.6	58.9	64.3	.47	-.03
13	+ 3	-75	0	3	71.1	59.7	65.4	.38	-.02
14	+ 3	-75	0	1.5	62.9	58.3	60.6	.28	±.01
15	+ 3	0	18	3	88.9	85.7	87.3	.57	±.01
16	+ 3	0	18	1.5	69.8	62.0	65.9	.65	+.03
17	+ 3	+75	18	3	84.6	73.4	79.0	.36	±.01
18	+ 3	-75	18	1.5	70.7	63.3	67.0	.36	+.02
19	- 3	0	0	3	61.4	51.7	56.5	.38	±.01
20	- 3	0	0	1.5	52.0	64.5	58.3	.34	+.04
21	- 3	+75	0	3	55.9	62.8	58.2	.29	±.01
22	- 3	+75	0	1.5	46.3	49.6	47.9	.35	+.02
23	- 3	-75	0	3	47.2	42.8	45.0	.21	+.02
24	- 3	-75	0	1.5	40.0	46.0	43.0	.25	-.01
25	- 3	0	18	3	85.2	72.0	78.6	.47	±.01
26	- 3	0	18	1.5	69.4	70.9	70.2	.46	+.02
27	- 3	+75	18	3	74.6	79.7	77.2	.25	±.02
28	- 3	-75	18	1.5	67.3	50.0	58.7	.23	+.01
29	0	0	0	3	90.6	81.7	86.2	.90	-.14
30	0	0	0	1.5	85.4	73.4	79.4	.86	-.12
31	0	+75	0	3	88.3	80.6	84.5	.74	+.05
32	0	+75	0	1.5	86.0	77.3	81.7	.74	-.21
33	0	-75	0	3	89.3	86.7	88.0	.63	-.14
34	0	-75	0	1.5	68.5	61.9	65.2	.63	±.20
35	0	0	18	3	93.8	95.3	94.6	.66	+.09
36	0	0	18	1.5	85.0	85.0	85.0	.79	+.05
37	0	+75	18	3	95.3	91.8	93.6	.58	-.05
38	0	-75	18	1.5	75.3	79.5	77.4	.55	-.05
39	+ 3	+75	18	1.5	77.8	77.8	77.8	.46	+.01
40	+ 3	-75	18	3	86.0	73.3	79.7	.32	±.01

Field Trial Number	Transmission Medium	Total Round-Robin Path Length, Miles	Terminals	Average of One List Each 6 Talkers	Average of 20 SCI Readings	Maximum Deviation of any SCI Reading from Average
IA	Microwave	24	Andrews AFB ⟷ Brandeywine, Md	96.9	.97	±.01
IB	HF SSB Radio	5300	Andrews AFB ⟷ Azores	77.5	.69	±.19
IIB	HF SSB Radio	5300	Andrews AFB ⟷ Azores	76.0	.64	±.20

Note 1 - All noises were white noise, 4400 cps low-pass filtered (-36 dB/octave) except those at a 0 dB S/N ratio. The masking source at 0 dB ratios was speech (continuous discourse by a single talker [C.W.]).

Note 2 - Low-pass filters had -36 dB/octave slopes above cutoff frequency.

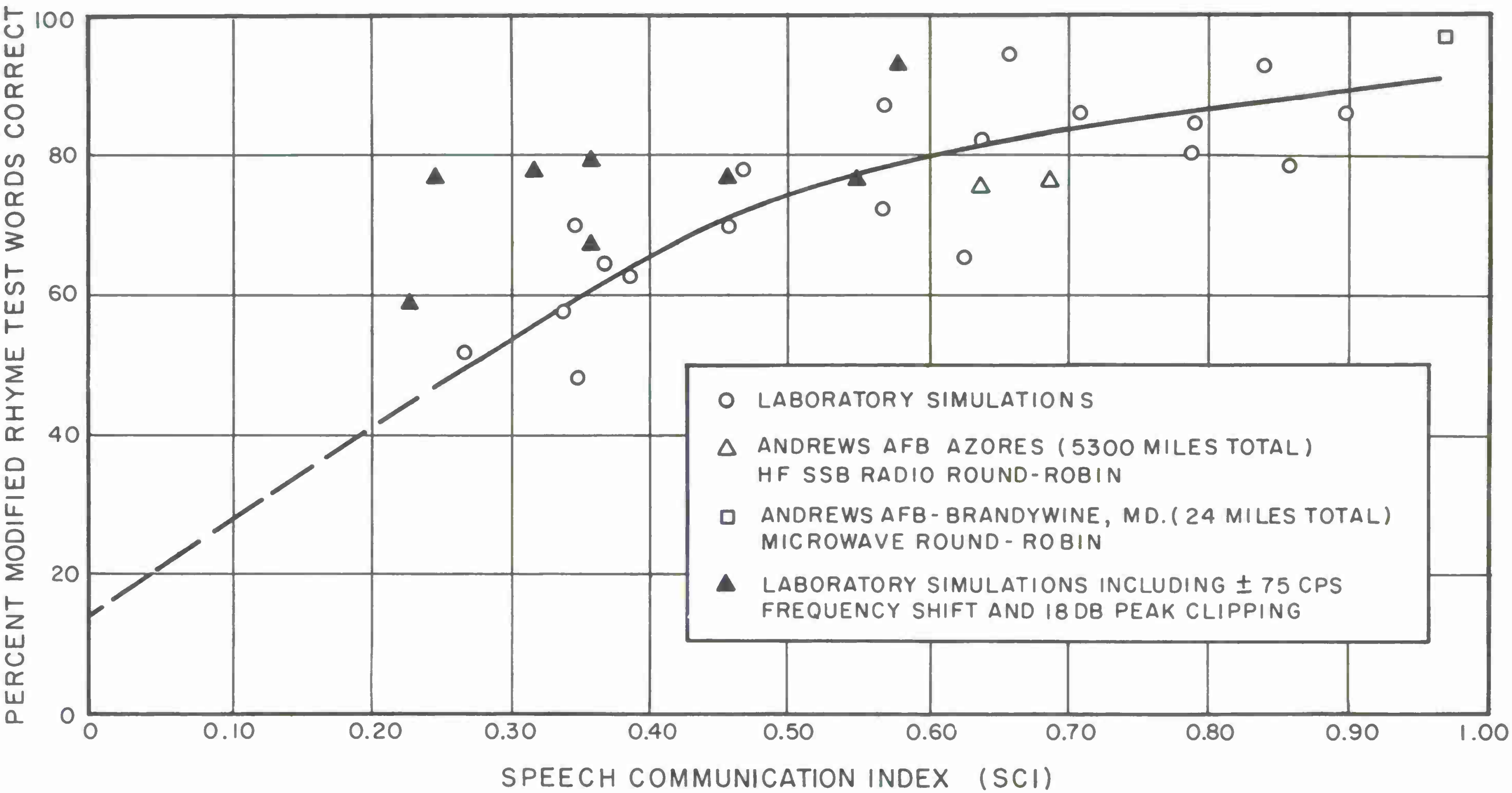

FIG. 15 SHOWING PERCENT MR TEST SCORES AS A FUNCTION OF SCI FOR LABORATORY SPEECH SYSTEMS, AF H-F SSB, AND MICROWAVE RADIO SYSTEMS. (SEE TABLE 1)

It is not possible to determine, from these tests, the source of the "error" causing this range of differences -- whether it lies with SCIM or with the speech tests. The speech tests themselves have an unreliability, for the number here administered, not much smaller than the range of differences found between SCI and measured intelligibility.

However, SCI can be expected to represent only an approximation to what the true capability of a system is for transmitting speech. For one thing, the use of only nine filters in the analysis set does not provide the fineness of spectrum analysis that would be necessary to insure extreme accuracy, and the other characteristics of the SCIM system (designed to correct or account for, in addition to frequency-amplitude irregularities in a transmission system, spread-of-masking effects, frequency shift, and amplitude distortion) represent somewhat idealized engineering averages for evaluating the effects of the variables involved.

All in all, it would appear that, except for certain conditions involving frequency shifts in excess of 60 cps or so when combined with peak clipping, SCIM gives an index measurement that can be used for evaluating a wide variety of speech systems and listening conditions that is nearly as "accurate" as calculated AI and/or measured percent of several MR speech tests administered to a crew of 8 trained listeners.

SECTION 5

FUTURE DEVELOPMENTS

We are planning to make certain additions to SCIM in order to make it a more general and useful instrument:

1. A provision will be provided in future instruments for a digital printout so that a printed record of the SCI value, time of day, date, etc., will be provided either with, or remotely from, the SCIM analyzer unit.

2. Acoustic couplers ("artificial heads") will be developed for use with SCIM. The "heads" will be such that a microphone or headset, earphone or loudspeaker can be attached to or placed near the head as would be the case when the system was being used by a person. The "head" at the transmitter end of the system would provide the SCIM signal acoustically to the microphone or mouthpiece of the system being tested and the "head" at the receiving end would pick up by a microphone in an artificial ear on the "head" the received signal for analysis purposes. When the artificial heads are used, the SCIM will, of course, measure the total system ambient noise conditions at both the microphone and earphone or loudspeaker positions, as well as the characteristics of the microphone and earphone.

3. Provisions will be provided so that SCIM can be used either in its present form, that is, supplying the input and output electrically to the system and/or acoustically; thus one can measure the performance of the system, including the microphone, earphone and ambient noise conditions independently of the electrical transmission and noise characteristics of the system.

4. Certain modifications are being contemplated that will permit SCIM to operate in time in a more automatic fashion than now possible.

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13. ABSTRACT Two major theories have emerged that attempt to relate in a quantitative way the physical characteristics of a speech communication system and the intelligibility of speech as perceived by a crew of trained listeners.

The more widely used theory is the basis for the "Articulation Index" (AI) formulated in 1947 by N. R. French and J. C. Steinberg of the Bell Telephone Laboratories. This theory holds that the signal (speech) -to-noise ratio (in 20 narrow frequency bands) of a communication system will, when properly weighted and summed, provide an AI value that is directly related to the intelligibility of speech heard over that system.

A second theoretical approach to this problem is embodied in the Pattern Correspondence Index (PCI) machine proposed by J. C. R. Licklider in 1956. This instrument determines the correlation in the frequency and time domains between the speech input to a communication system and the speech output of that system. The subject of the present paper is an electronic device called SCIM (Speech Communication Index Meter) that is designed primarily in terms of the Articulation Index theory; however, SCIM accommodates additional factors affecting speech that are not involved in the original procedures for calculating AI. The SCIM signal generator generates a 3-second signal burst, which is fed to the electrical input of the system under test; within 12 seconds the SCIM analyzer, located at the receiving station, calculates and displays an index, ranging from .00 to .99, which reflects the ability of that system to transmit intelligible speech.

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